





درس ۲۶

مروری بر فیلترهای وفقی

Overview of Adaptive Filters

کاظم فولادی دانشکده مهندسی برق و کامپیوتر دانشگاه تهران

http://courses.fouladi.ir/dsp

Overview of Adaptive Filters

Digital Signal Processing

The Filtering Problem

• Filters may be used for three information-processing tasks



- Given an optimality criteria we often can design **optimal filters**
 - Requires a priori information about the environment
 - Example:

Under certain conditions the so called Wiener filter is optimal in the mean-squared sense

- Adaptive filters are self-designing using a recursive algorithm
 - Useful if complete knowledge of environment is not available a priori

Applications of Adaptive Filters: Identification

• Used to provide a linear model of an unknown plant



- Parameters
 - *u* = input of adaptive filter = input to plant
 - y = output of adaptive filter
 - d =desired response = output of plant
 - e = d y = estimation error
- Applications:
 - System identification

Applications of Adaptive Filters: Inverse Modeling

• Used to provide an inverse model of an unknown plant



- Parameters
 - u = input of adaptive filter = output to plant
 - y = output of adaptive filter
 - d =desired response = delayed system input
 - e = d y = estimation error
- Applications:
 - Equalization

Applications of Adaptive Filters: Prediction

• Used to provide a prediction of the present value of a random signal



- Parameters
 - u = input of adaptive filter = delayed version of random signal
 - y = output of adaptive filter
 - d = desired response = random signal
 - e = d y = estimation error = system output
- Applications:
 - Linear predictive coding

Applications of Adaptive Filters: Interference Cancellation

• Used to cancel unknown interference from a primary signal



- Parameters
 - u = input of adaptive filter = reference signal
 - y = output of adaptive filter
 - d =desired response = primary signal
 - e = d y = estimation error = system output
- Applications:
 - Echo cancellation

Stochastic Gradient Approach

- Most commonly used type of Adaptive Filters
- Define cost function as mean-squared error
 - Difference between filter output and desired response
- Based on the method of steepest descent
 - Move towards the minimum on the error surface to get to minimum
 - Requires the gradient of the error surface to be known
- Most popular adaptation algorithm is LMS
 - Derived from steepest descent
 - Doesn't require gradient to be know: it is estimated at every iteration
- Least-Mean-Square (LMS) Algorithm

$$\begin{pmatrix} \text{update value} \\ \text{of tap - weight} \\ \text{vector} \end{pmatrix} = \begin{pmatrix} \text{old value} \\ \text{of tap - weight} \\ \text{vector} \end{pmatrix} + \begin{pmatrix} \text{learning -} \\ \text{rate} \\ \text{parameter} \end{pmatrix} \begin{pmatrix} \text{tap -} \\ \text{input} \\ \text{vector} \end{pmatrix} \begin{pmatrix} \text{error} \\ \text{signal} \end{pmatrix}$$

Least-Mean-Square (LMS) Algorithm

- The LMS Algorithm consists of two basic processes
 - Filtering process
 - Calculate the output of FIR filter by convolving input and taps
 - Calculate estimation error by comparing the output to desired signal
 - Adaptation process
 - Adjust tap weights based on the estimation error



LMS Algorithm Steps

• Filter output

$$y[n] = \sum_{k=0}^{M-1} u[n-k] w_k^*[n]$$

• Estimation error

$$e[n] = d[n] - y[n]$$

• Tap-weight adaptation

$$w_k[n+1] = w_k[n] + \mu u[n-k]e^*[n]$$



• The LMS algorithm is convergent in the mean square if and only if the step-size parameter satisfy

$$0 < \mu < \frac{2}{\lambda_{\max}}$$

- Here λ_{max} is the largest eigenvalue of the correlation matrix of the input data
- More practical test for stability is

$$0 < \mu < \frac{2}{\text{input signal power}}$$

- Larger values for step size
 - Increases adaptation rate (faster adaptation)
 - Increases residual mean-squared error